462dsp/FM

4-bands digital audio processor

User's manual

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About this manual

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What's in the box?

You must find the following components into the box.

- *1 Solidyne 462dsp processor*
- *1 user's manual*
- *1 AC Interlock*
- *1 CD-ROM with VirtualRack 5 software.*
- *1 Guaranty certificate*
- *4 self-adhesive rubbers tops*

Please check when receive to verify that all components are okay.

About the montage

462dsp processors are designed for rack montage in standard racks of 19", requiring 2 rack height units. If you prefers, the processor can be placed on a table. Four self-adhesive rubber tops are given for this purpose.

When mounting, start fitting the bottom screws, and soon the superior ones. Use flat screws with flexible o-rings (rubber, PVC, etc.). The frontal panel is made of aluminum, so you must take special care of don't to apply excessive force on the screws since it can cause deformation or even break the angles of the panel.

ADVICES

AC Voltage	The unit cans wok with **110 or 220 VAC.** A voltage switch on the rear panel selects the correspondent voltage. **ALWAYS CHECK THIS SELECTOR BEFORE PLUG IN.**

In order to reduce the risk of electrical shock, do not retire the covers of the cabinet. The internal pieces do not require maintenance of the user. Refer the technical maintenance to qualified personnel.

	The power cord provided with the unit gives Earth return to the processor. Do not replace it nor uses adapters. **MAKE SURE THAT HAVE WITH A GOOD GROUND TAKING.**

	The exclamation icon within a triangle that appears in this manual is for alerting to the user about the presence of important instructions on the operation and maintenance of the equipment.

	Letter "i" within a circle that appears in this manual is for alerting to the user about the information; advices and tips of extreme importance.

1.1 Basic connection and settings

CONECTIONS

Power supply

Before plug in; check the position of the AC VOLTAGE selector, on the rear panel (200-240 V, 50/60Hz; or 100-130 V, accords to correspond).

Inputs

Analogical audio inputs are balanced, using XLR (female) connectors. Connect the PROGRAM OUTPUTS of your console to these inputs. Take care of not invert the phase of the channels.

Optionally; the 462dsp has digital AES-3 (AES/EBU) input, for digital consoles or links. This input supports 16/24 bits and sample rates from 30 up to 96 KH. When the digital input is used, we recommend connecting the analogical inputs too. 462dsp will switch automatically to these inputs in case of failure on the AES-3 signal.

Outputs

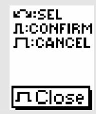
Connects the 462's MPX output (BNC) to your transmitter exciter.

Power on

The unit has an ON/OFF switch on the rear panel. The main screen Hill show a welcome splash on start up; and then will load the last used program.

SETTINGS

Controls

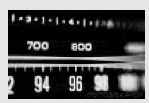


All settings values are entered using a JOG Wheel. Turn the JOG to select options and to change values. Pressing briefly the JOG, confirms an option. Pressing and holding the JOG (around one second), go to the main menu. Each screen graphically indicates the actions available with the JOG (see figure).

Input

The default input level is +4 dBu. If your console manages a different level, you must adjust the processor input level.
Press briefly the JOG to enter to the main menu. Choose the option "Setup Input" and adjust the level according to the nominal specified on your console.

Programs

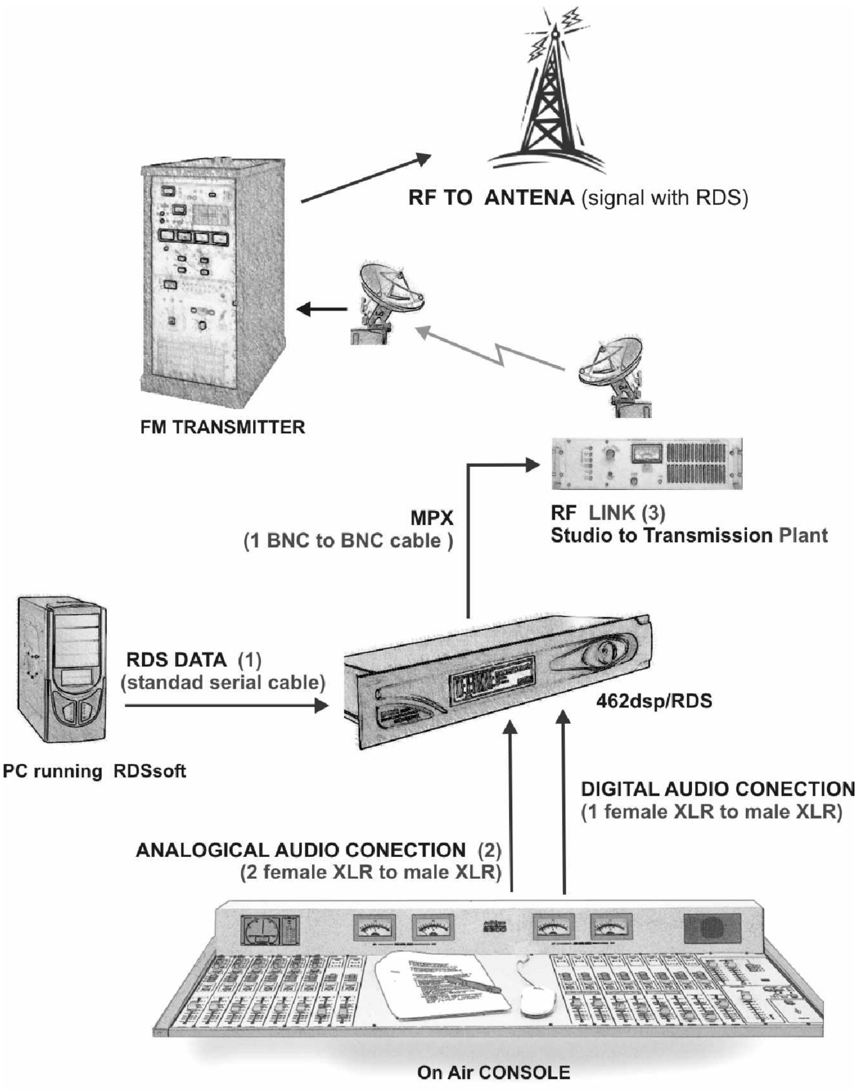


Your radio already is on the air with the sound of the Solidyne 462dsp. At this point, you must be anxious to listen what 462dsp is able to do with the sound of your radio on the air… to begin we will explain briefly how to select the factory presets. Tunes your radio in a good audio equipment and do the following:

Turn the JOG. Note on the screen how the different presets programs names changes. Pressing briefly the wheel load the program, changing the processing and the sound on the air.

The 15 PRESET PROGRAMS can not be changed. You have 15 USER MEMORIES to create your own adjustments. You can start copying a preset program to a user memory, and soon modifying it.

1.2 Brief diagram for general connections



1. If your computer does not have serial port; you can use a **RS232 to USB** adapter to connect the 462dsp on a USB port.

In units with option RDS, you need to connect a computer to the processor for RDS programming. This connection can be not permanent (in case that a fixed text is transmitted; it is stored into RDS memory).

2. When the processor connects to the console using the digital inputs, it's convenient to make the analogical connection too. In case of fault in the digital signal, 462dsp changes to the analogical inputs automatically.

3. In this example a RF connection was used to transport the MPX signal from the processor to the transmitter, which assumes installed in a build far from the studies.

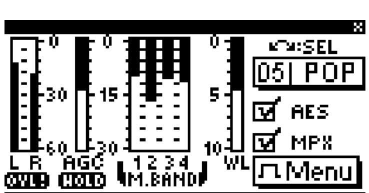
In other configurations, the processor locates in the transmitting plant and a digital link is used to send the audio (multiple sends can be used for backup).

In this case, the remote control of 462dsp can be implemented through a data channel of the link that allows sending RS-232 data.

1.3 Understanding the main screen

Next you will see an overview on the 462dsp main screen. Later each processing stages and its settings screens are fully analyzed.

When the unit starts up; a boot screen appears showing the current version of the operating system. Once started, the unit presents the main screen, which is described briefly at next as introduction to the 462dsp graphical environment.



The progressive bars show the action of several processor stages. The name of the current program and state of inputs and outputs also appears on this screen. From left to right you find:

Ø The **input VU meters** (L & R). Below the vu-meters an overload **"OVL"** warning can appear, which turns on 8dB before the "digital clipping" level (@ +18dBm). This warning turns off last 5 seconds if the "clipping" condition disappears.

Ø The following bar shows the action of AGC (Automatic Gain Control) that fits the input gain so that the signal arrives at the processing stages with constant level. In others words; if the signal from the console arrives with low level (or very high), the processor automatically compensate its input gain so that the output level is uniform. This is a gated AGC. If the input signal falls abruptly,

the AGC holds its current value; to avoid an undesirable effect known as “breathing”, that appears when the signal is very low and the AGC compensates their gain excessively increasing the background noise in the voice intervals. In normal operation this indicator will have to work at middle scale (15 dB).

Ø The following indicators show the action of the multiband compressor. There are four compression bands called **loudness bands** (LF, M1, M2, HF). These indicators (like the AGC) show the containment of each band, that is to say, the compression (gain reduction) applied to the signal, for that reason they “grow downwards”.

Band LF corresponds to bass notes of the audio, below 160 Hertz, M1 and M2 to mid tones and HF to the highs.

The LOUDNESS concept associates to the sensation of sonorous power perceived by the human ear. With no need to exceed the maximum percentage of modulation fixed by the laws, the processor can obtain that the radio “sounds” stronger. The radio with better processing will be the one that better sounds in the dial.

As will be explained later, another objective of the processing is to increase the energy in the entire audio spectrum, to produce the maximum possible modulation in the RF carrier, which is translated in greater radiated energy, improving the transmission reach. For this, the processor applies a complex procedure of compression on each audio band, producing greater sensation of loudness to the ear.

The size of each luminous bar indicates the compression degree at every moment. Its effects, obviously, changes according to the different types of music and voice; but in general can say that when a band indicator begins to light (values up to 10 dB) the action is smooth and is totally accepted by the human ear. With values of 20 dB or greater will notice a greater “sonorous force”, being a level of extreme processing.

Another important value to consider is the **recovery time.** This is the time that takes a band in recovering its previous gain. This is appraised at first observing the indicators. The slower it is the recovery, the processing sound soft and the music sounds natural. However, when they are fast, an increase of the loudness takes place, but the sound can becomes more “rough”.

Ø “AES In” indicates if the unit is using the digital input. It activates just at the moment at which the digital data enter.

þ = digital input

¨ = analog input

Ø **“MPX Out”** indicates that the processor is using the internal FM stereo coder.

Ø Finally; the screen shows the name and number of the current program. You can explore the programs list turning the JOG; but a program is not active until you load it pressing briefly the wheel.

A **program** is a set of adjustments stored in an internal memory.

You access to the **MAIN MENU** by pressing and holding the JOG wheel. From here you can access to all functions and features of the processor.

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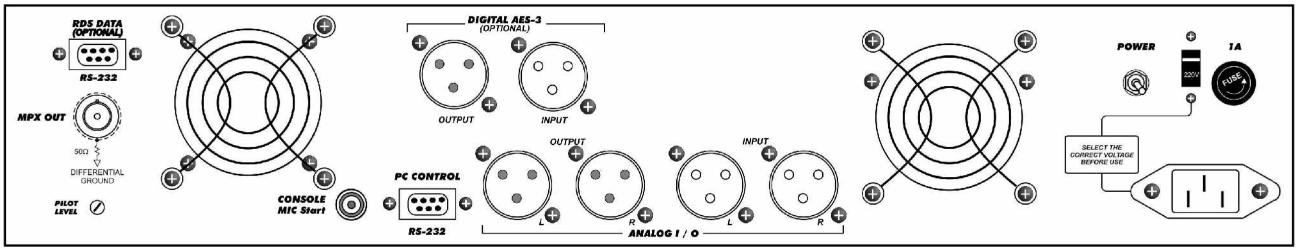


2.1 Power supply

- The AC voltage variation must stay smaller to 10%. Otherwise, uses an UPS.

⚠	Always CHECK the correct position of the VOLTAGE SELECTOR (200/240 V o 100/130 V, according to the country)

AC wires do not have to be mixed with audio wires, especially with analogical ones.

Remember that all audio installation must have a trustworthy grounding. We recommend accomplishing with the effective norms - Article 810 of the National Electricity Code (NEC); ANSI/NFPA Nº 70-1984 in USA; IRAM 2379 and 2281-3 in Argentina. This norm provides information and guidelines for a consistent grounding.

2.2 On mounting

- 462dsp can be mounted in a standard rack of 19"; or can be used on a table. Rubber tops are provided with the unit. Do not place the unit on unstable surface or shelf; the apparatus could fall, causing damages to someone and to be damaged the unit.
- The ambient temperature must stay between 5ºC and 40ºC. Avoid direct solar ray incidence on the processor or proximity of heat sources.
- The openings and grooves allow the circulation of air inside the unit. These openings do not have to be blocked nor covered, not to obstruct the refrigeration of the internal components.
- 462dsp has internal protection against RF fields, which allows locate it next to transmitters (AM or FM). Avoid strong electromagnetic fields (power transformers, big motors, etc).

2.3 Analogical audio connections

Inputs and outputs are electronically balanced. The inputs are "bridging" type, with impedance greater than 10 KOhms. The connectors used, as is standard, are female XLR-3 for the inputs and male for the outputs.

⚠	Take specially care with the phase.

- Use one pair shielded audio cables of GOOD QUALITY, preferably with double shielding. The maximum length recommended is 30 meters, although in special cases it is possible to achieve 100 meters accepting a little loss at high frequencies.
- The connection of this cables are made as is standard. See the following table:

Balanced input/output connections:

1 = GND
2 = balanced positive phase (+)
3 = balanced negative phase (-)

Unbalanced connection:

Inputs: Signal = 2;
Ground = joint 1 and 3
Outputs: Signal to pin 2; leave pin-3 unconnected.
Ground = pin 1

2.4 Digital audio connections

Optionally, the processor has digital **AES-3** inputs/outputs (models 462dsp/AES).

The **digital input** supports:

Resolution: **16** - **24 bits**
Sample rate: **30 KHz** a **96 KHz**.

Internally 462dsp works at 24 bits/192 KHz, the digital signal is converted internally by a stage *called "Resampler"*.

When the digital input is used, is convenient to connect the analogical input too. In case of losing the digital connection, the processor switches automatically to the analogical inputs. The input mode is selected from the main menu choosing the option "Setup Input", as is explained later.

S/PDIF:

You can connect an S/PDIF output to the AES3 input of 462dsp using an S/PDIF to AES-3 adapter. The figure shows a compact adapter XLR BNC.

AES-3 output has a resolution of 24 bits with a samplerate selectable between 48/96 KHz (see "3.2.2 – Output Setup"). The connector is male XLR.

AES3 input and output cables connect as following:

XLR	Signal
1	GND
2	AES3 (1)
3	AES 3 (2)

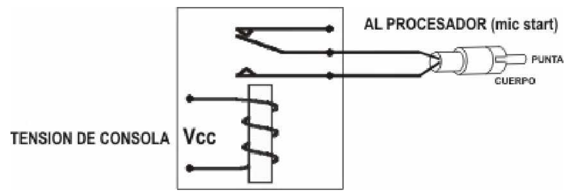
AES-3 standard connection

2.5 Console MICstart

With the 462dsp the current processing program can changes when the study microphones activates. In this way you can use a processing specially done for voices. The default adjustment for voices is the program "09: Voice Impact". Obviously; you can copy and customize this program.

The commutation takes place when grounding the MICstart input. When this happens the processor exchanges the program and the access from the JOG wheel is blocked. When the MICStart input is opened, 462dsp returns to the previous program.

"Console MICstart" uses a connector RCA type. With consoles *Solidyne 2300 series*, MICstart is connected directly to the *Digisolid* output of a microphone channel (please refers to the manual of the console). In other consoles, the control can be solved using the tally signal to activate a relay.



Connect the relay contact directly to the RCA "MIC Start"; and the coil in parallel to the tally light.

⚠	The relay coil voltage depends on the voltage gives by the console.

When the relay receives tension, the contact is closed activating the Mic Start function in the processor. The voice's program remains active while the contact is closed.

This feature comes disabled from factory. In order to use it, you must enable the option "MIC PROCESSING" in the processor setup (see "3.2.3 - Processor Setup").

2.6 MPX output

The MPX cable will be a RG-59 (coaxial 75 ohms), like the used for CATV. The output connector is BNC type. The maximum length recommended for this cable is 25 m. Take care with the grounding; although this rarely is cause of problems because all Solidyne processors have MPX differential outputs, that is to say, with the Earth isolated of the cabinet, to avoid ground loops.

If some residual humming appears when the system is on the air; power off the processor. If the humming disappears, check the input connections at the processor. If, however, the humming continues (and only disappears unplugging the MPX cable), this indicates some problem with the grounding.

When enters to the transmitter through the MPX input, make sure that the internal preemphasis network IS DISCONNECTED (that is to say, flat response from 20 to 100 KHz). Contrary, when use an external stereo coder, make sure that the generator INCLUDES the preemphasis curve. This is thus since the 462dsp audio output DOES NOT INCLUDE preemphasis (only the MPX output has preemphasis).

2.7 RDS CODER (OPTIONAL)

Models 462dsp/RDS have **internal RDS coder**.

RDS (Radio Data System) is a system developed by the European Broadcasters Union (EBU). It allows adding to a conventional FM transmission, additional information by means of the inclusion of sub-carrier that contains data.

Their main applications are:

1. The automatic tuning of the receiver to a radio network selected by the user, which, allows to listen to a program, for example Classic Radio, during a long trip by the route, with no need to tune manually the receiver to another station of the same radio network, when the reception happens to be deficient when leaving the zone on watch of a determined station.
2. Show on the receiver screen the radio network name that is tuned, for example Radio 1, and the kind of program: the general news, talk-show, sports, music, varieties, monk, etc.
3. The automatic reception of information related to the traffic. When this feature is selected the news has priority on the traffic, so that the receiver will exchange, automatically, within a same network, to the transmitter that emits information on the traffic, and once finished this information it will return to be in tune, automatically, the transmitter that previously was selected.

2.7.1 RDS – PC connection

For setup and control of RDS stage; the 462's "RDS Data" port must be connected to a computer, using a standard COM (RS-232).

RDS Data	PC
2 (RxD)	**3 (TxD)**
3 (TxD)	**2 (RxD)**
4 (DTR)	6 (DSR)
5 (GND)	**5 (GND)**
6 (DSR)	4 (DTR)
7 (RTS	8 (CTS)
8 (CTS)	7 (RTS)

Use a standard crossed serial cable (as known as "null-modem"). Two female DB-9 connectors are required. The following table shows the diagram of connection, although only the connections emphasized in bold are needed.

The computer is used to transmit data to RDS coder. These data can change in real time, like when the names of the songs are transmitted, or can be a fixed text that it is stored in the internal memory of the RDS coder.

In order to command the RDS coder; the software *Solidyne-Magic RDS* is required. This application is included in the 462dsp CD-ROM, provided with the unit. Using this tool you will be able to start to send RDS data. Please refer to the RDS help file for detailed information on the software.

2.7.2 Connecting to the transmitter

The models 462dsp/RDS do not require a special connection. MPX signal contains the RDS information that is injected directly to the transmitter when connecting the MPX output

The digital signal that it contains information RDS, is transmitted with a speed of 1187,5 bit/s and modulates a subcarrier of 57 KHz, using the method of Amplitude Modulation with suppressed carrier, that is added to the multiplexed stereophonic signal; that it is sent to the transmitter input. See the blocks diagram shown in "Chapter 1".

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3.1 Basic actions

3.1.1 Using the JOG wheel

All the installation settings and each parameter of audio processing are managed with the JOG wheel into a friendly graphic interface.

The frontal panel presents a big blue screen of dot matrix, and a rotating control (JOG) with push button. Their use is very simple:

- Turn the JOG control to select an option or to change values (i.e.: changing levels; select Yes/No; etc.)
- To confirm an option/value **press the JOG briefly**. It's like make "click" with the mouse.
- **Push and hold the JOG by a second** to enter to the Main Menu or to cancel a screen. It's the equivalent to double-click with a mouse.

3.1.2 Processing programs

462dsp has **15 presets** programs and **15 user** memories. Programs from 00 to 14 are programs made by the engineers of Solidyne, **ready to be on the air**. There are different adjustments for different music's styles. Each program has a name that identifies it (Jazz, Rock, Pop, Melodic, etc.) but the **names are only indicatives**; all music styles sounds well with any preset. Processing programs are described with detail later.

You can customize the programs copying it to the user banks (15 to 29) and edit them. By default, user memories are "empty" (strictly speaking with "flat" processing adjustments). You can create a new adjustment changing the values of each parameter on the user memory; or copying a preset to the user memory (recommended). See "4.3.1 – Copy programs"

3.1.3 Modes of control

The processor can be controlled of many ways:

a) Can be manually controlled from the frontal panel.

b) The programs can be chosen automatically from the OnAir PC. In this way you can use a specific processing for each song or musical style.

c) The programs can be created and edited using the **462dsp VirtualRack software**. This software allows saving the 462's programs in the Hard Disk (files .462) that you can send by mail, so that other radios of your chain share the customized adjustments. In addition; this software allows creating a **scheduling programming**, to change the processing program at certain hours from the day. Virtual Rack runs in background using very few resources. In fact this software can run on the On-Air computer.

d) Connecting the PC to Internet, you can controls the software via "Shared desktop" Windows feature.

3.1.4 Access password

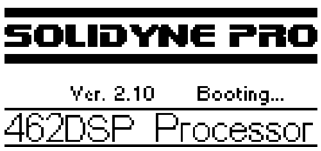
You can define a password of 3 characters (letters, numbers and signs) to avoid that non authorized people make changes on the programming adjustments.

Each time that you try to access to the main menu the processor will ask you for the password.

Note that **you don't need the password to change the current preset on-air**. Please see "3.2.3 - Processor Setup".

3.2 INSTALLATION SETTINGS

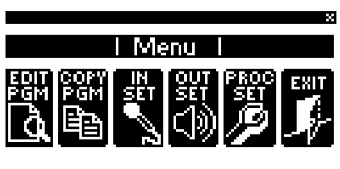
a) On start up, the 462dsp shows by a few seconds the boot screen indicating the firmware version:



Once started; the processor present the main screen; described in the item 3 of the Chapter 1.

i	The unit always starts loading the last used program.

b) Pressing and holding the JOG you accede to the **Main Menu.**



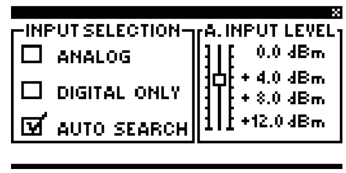
Once into main menu; turn the JOG until select the desired option. The selected option is marked by **a little arrow** that appears **below the icons**. Touch briefly the JOG to confirm the selection.

WARNING
Remember that being in the main screen, turning the JOG or touching it briefly, you accede to the mode 'program selection'. In this mode; turning the JOG you explore the 462's programs. In order to leave the 'programs selection' mode, press and hold the JOG. In order to accede to the main menu press and hold again.   **If being in the main menu or any settings screen changes not occur during 10 minutes, the unit returns to the main screen, discarding the changes.**

3.2.1 INPUT SETUP

a) Here you define the **audio level** and **type of input**. The operation on this screen follows the concept of the previous ones: to select an option ("Input selection" or "Input level") turn the JOG and touch briefly to confirm. Now turn the JOG again to change the value of the chosen option. Touch briefly to confirm.

Selecting this option the following screen will appear:



INPUT SELECTION

¨ **Analog:** Select this option when the processor is connected ONLY through the analogical input.

¨ **Digital only:** Choose this option when the unit is connected ONLY through AES/3. Also the analogical inputs can be connected, as back up, but they will be used only in case that the carrier of signal AES/3 interrupts. In this case, the equipment exchanges automatically to the analogical inputs, being in that mode until the digital signal is reestablished.

þ **Auto Search:** (default) It uses the AES/3 digital inputs in case of finding a source of digital signal. Otherwise, uses the analogical inputs. In case of finding only silence or a total lost of digital AES/3, the unit switches to the analogical inputs. This type of faults in signal AES/3 is characteristic when "a digital data link falls". As signal AES/3 is regenerated in the receiver, the digital streaming follows present, but without content of audio (channels muted).

Reestablishing the signal in Auto Search mode: After losing AES/3, or in absence AES signal, or absence of audio in this digital signal, the equipment switches automatically to the analogical mode. The reestablishment to the digital mode happens in immediately in case signal AES/3 is had completely lost. Since when recognizing a valued digital carrier, the processor quickly returns to use the digital audio like source of incoming signal. In case the digital signal exists, but the same one contains silence, the processor operates as follows: At intervals of 5 minutes the processor interrupts briefly (1 sec) the audio on the left channel, connecting in its place the digital signal of the left channel; verifying soon if the digital signal were reestablished. If it continues in absolute silence, then it reestablishes the analogical signal. This process is repeated time and time again until as returns signal AES/3. When the digital audio reappears, the processor returns to digital mode.

While the digital signal remains absent, the audio on the left channel is briefly muted at intervals of 5

minutes, to alarm to the engineers of the radio (and to the operators) that the processor is working with the analogical backup inputs. This has been intentionally designing in this way, since otherwise; an accurate confirmation does not exist on problems in the signal AES/3.

A. INPUT LEVEL

Define the nominal level of the analogical input signal. So that the AGC (automatic gain control) operates correctly, is very important to fit this level correctly.

This value must agree with the **nominal output level of your console** (in the case of Solidyne consoles this level is + 4 dBm).

NOTE: *If you unknown the output level of your console, put the console's level at 0 VU peak and reads the level in the 462dsp's input VU-meters.*

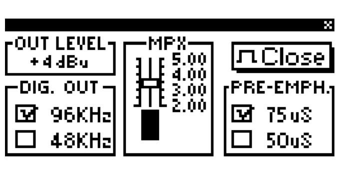
With the adequate input level, the AGC works around 10 to 15 dB of gain reduction. This can be seen in AGC indicator who will have to indicate 15 dB with strong program signal, lowering to 8 dB or less when the signal reduces. If the program signal level abruptly falls, the indication "HOLD" appears.

Overload warning (OVL): appears in the main screen when the input signal reaches +10 dBm; that is to say, +8 dB before the 'digital clipping' threshold (placed at +18 dBm). The indication disappears last 5 sec if the clipping disappears.

The digital input fulfills recommendation AES-K12, applied to broadcasting, which takes as reference for 0 VU a digital input level of - 12 dBFS.

This implies 12 dB of headroom. The user cannot modify the input level when digital input is used.

3.2.2 OUTPUT SETUP



OUT LEVEL: It refers to the analogical output. Level is fixed at +4dBm.

DIG. OUT: Define the sample rate of the AES-3 output. It can be 96 KHz or 48 KHz.

MPX Level: Define the MPX output level. It can change from 2V to 5V peak to peak.

You must set the value according to the value that the transmitter needs to reach the 100% of modulation (75 KHz deviation).

PRE-EMPHASIS: Allows to adjust the pre-emphasis curve according to the regulations of your country (I.E.: Europe = 50uS; USA, ASIA, Latin-American =75uS)

3.2.2.1 Adjusting 100% of modulation in FM

For this adjustment use a program material of high density; with voice and music (i.e.: commercial spots). Select the program "08:MaxLoudness" (it produce high output level). Next change the MPX output level until obtaining 75 KHz of deviation, measured with a Monitor of Modulation (like the Solidyne VA16) or with the modulator meter of the transmitter. For fine adjustment of the 100% you can use the input gain control of the transmitter's exciter.

In the daily use of the processor, probably the indicators of modulation of needle type overpass the 100%. This can be due to over-impulses of the ballistics or to that it responds to values average of sine wave and the indication is erroneous with the high processed audio. Please see the following notes.

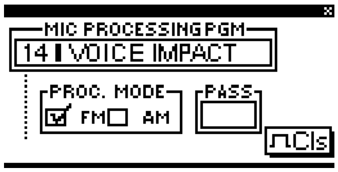
3.2.2.2 Notes about FM

Remember that many countries follow the recommendations on modulation of the FCC (USA). Recommendation 73268 indicates that the FM must stay all the high that is possible, but without exceeding 100% in peaks of frequent recurrence *("In no case is it to exceed 100 % on peaks of frequent recurrence")*. This indicates that in transitory peaks (and no frequent) 100% of modulation can be surpassed staying within the legal frame.

The 462dsp processor is designed to fulfill this **FCC Norm**, being allowed to surpass 100% in no recurrent peaks slightly. When fitting the modulation, verifies the norms of its country.

When making the programs; take in count if there is an UNIQUE program for the whole day. In that case commitment solutions will be use to accept all types of voices and music that the Radio emits. Using the remote control from PC, diverse programs can be created. This eliminates the commitments, so each program will be the optimal for that type of music or voice. In addition, this automation eliminates the listener's ear fatigue associate with the radio stations that use rigid processors.

3.2.3 PROCESSOR SETUP



MIC PROCESSING PGM: MICstart input allows to change the processing program when the operator activates the microphones at the console, applying a specific processing to obtain high impact voices.

This is a very useful feature, since it's not possible to obtain an adjustment that sounds optimal for music and voice simultaneously; for being two audio signals morphologically different. The attack and recovery times of the AGC differ sensibly for both cases; and must be carefully fit to avoid that its action is audible.

This option is **disabled from factory** (program 30:Mic processing off). In order to enable it, select "Mic processing PGM" and choose a program created for voices (for example 09:VOICE IMPACT).

The commutation controls from a connector in the rear panel, detailed in the installation chapter (see "2,5 Console MIC Start").

TIPS

The **programs for "music" and "voice" do not have to be radically different**. The gains of the density EQ and the AGC output must stay without greater variations. The differences will be fundamentally in the attack and recovery times of the AGC and the multiband compressor.

It agrees to create the adjustment for voices starting from the adjustment chosen for music. The attack and recovery times used for music usually are slower than those required for the voices, mainly in the AGC. Observe and compare the factory presets.

PASSWORD: You can define a password to avoid the access of unauthorized personnel to the main menu to change the settings. Once defined, 462dsp will ask for the password whenever you try to accede to the main menu, but you don't need the password to explore the program list and change the program on-air.

The password must have 3 characters. Any combination of letters, numbers and symbols can be used.

- To enter a password, go to field "Password" turning the JOG and touch it briefly to confirm.
- A flashing cursor appears to enter the first character. Turn the JOG until reach the wanted symbol and confirm with brief touch. The cursor will jump to the next character.
- Proceed in the same way to choose the others characters.
- After confirm the last character, the option "OK" is on focus. PRESS AND HOLD to confirm the password. A brief touch in this point will make jump the cursor to the first character again.
- **Press and hold the JOG cancels the process** in any instance. (equivalent to escape), **except when "OK" is selected**, in such case confirm the password!
- To **delete the password**, edit it entering 3 space characters.

Write down the password in a secure place, in order to do not loose it. If you forgot the password, please contact with Solidyne; and y you will receive instructions to restore it.

PROCESSOR MODE: Shows the processor type. The unit can be prepared for FM or AM stations. This is defined in factory, because requires some internal changes in the circuit board; as well as a different programming of the flash memory with the corresponding constants for the AM band.

4.1 PROGRAM SELECTION

The unit has 15 default programs (00 to 14) and 15 user memories.

The programs selects from the main screen. Turn the JOG to explore the list and touch briefly to put a preset on the air.

- Note that, while you explore the programs, over the program name appears the navigation instructions.
- While these instructions are on screen, you are in the mode "exploring programs".
- Selecting a program this legend disappears.
- Touch and hold the JOG to cancel the exploration. The name of the current on-air program returns on screen.

THE CHANGE WILL TAKE EFFECT ON THE AIR JUST WHEN YOU LOAD IT BRIEF TOUCHING THE JOG.

4.2 DEFAULT PROGRAMS

462dsp has 15 programs created by the Solidyne engineers. Next each preset is explained. These adjustments are the base for the creation of new programs.

00: JAZZ – CLASSIC (orchestral)

This is a "soft" program. The objective was respect the original equalization and balance of the orchestra; maintaining its dynamic expression; by all means: within the limits of the FM transmission. For this the AGC uses slow attack and recovery times, and its output level is around 4 dB, for not to produce excessive compression. Remember that an excessive compression will cause a spectral imbalance unacceptable for Jazz music and orchestras.

The times for the bands M1 and M2 require special attention, since there is much participation of soloist's instruments in this music. The recovery times must be similarities, otherwise timbre modulations will take place, mainly in wind joints.

This adjustment reinforces the lows subtly. The dynamics of the bass stays using fast recovery times for the low band (LF).

The high frequencies are not emphasized. The objective was to prioritize the fidelity; "clearness" and definition of the instruments over the "brightness effect". By such reason the recovery time of the high band (HI) is relatively slow. Consider that the listener always can emphasize the trebles in his tuner by equalization.

This program also is very appropriate for TANGO orchestras, although generally the tango tolerates a little more processing (AGC OUT = 7 dB) and can require adjustments in the times of the bands of M1 and M2, by the stylistic differences in the use of the vocal intervention respect to the Jazz.

01: JAZZ IMPACT

It's similar to the previous one; but a little more "strong" (slower attack times and faster recovery times). JAZZ IMPACT apply a little increasing to the BASS CLIP threshold to reinforce the bass "punch". This processing works very well with Jazz-POP, country and blues, folklore and tango.

02: MELODIC

Here was prioritized the human voice over the musical support. To respect the "color" and shades of the soloists was the central objective when creating this program. As reference was used material of diverse soloists of ballads, boleros and melodic music. Although this adjustment is centered in the voice, is not equivalent to adjustment "VOICE IMPACT", since the characteristics of the sung voice differs remarkably from the spoken voice.

The multiband processing is moderate. The AGC output is about 4.3 dB. Greater values will cause very audible changes of timbre in some soloists, due to the imbalance that take place between the bands with excessive compression. The attack and recovery times stay relatively slow, not to lose the dynamic profiles completely.

The mid bands M1 and M2, where the vocal energy is concentrated, apply a considerable compression to the voice, since their attack times are fast (17 mS) and their recovery times are slow (540 mS and 181 mS respectively).

The adjustments of the density EQ compensate the balance between bands to obtain a flat response, comparing the processed material with the original one. For that reason band M1, that is the one that receives greater compression in average, is emphasized not to lose presence the mid-low.

03: MELODIC HI BOOST

This adjustment is similar to the previous one, but can be said that a little "hardest". The high presence was increased, releasing the attack (18 mS) and lowering the recovery time (60 mS). The mid-low band (M1) has a faster recovery time (400 mS) and the low band is slightly more compressing. The output level of the AGC was increased to 5 dB. A bass punch is added setting the "Bass Clipper" control at -4 dB.

04: MELODIC IMPACT

This adjustment is similar to the previous one, but with al little hard processing.

05: ROCK

This adjustment prints a remarkable processing to music. The gain of the AGC is around 9 dB, which that considers a "strong processing". The "force" of this preset is sustained not only by the AGC output level, but also by the fact that all the bands have slow attack times and fast recovery times. By such reason, the fifth band of highs post-processing will have noticeable incidence on the processing.

When all bands have fast recovery times; the action of the attack times prints "force" to the sound, given "punch" to the music. This is because being fast the recovery, the next impulse finds "released" the compressor, and this will take time again in containing the signal; which lasts the attack time. This effect is more notable in LF, and as it is explained in the POP

adjustment, one of the keys to obtain forceful “pumping” bass drums (or beats for techno music).

The density EQ reinforces the lows slightly. The marked attenuation in highs looks for to control the ear fatigue produced by the electrical guitars, which can present high distortion (intentionally applied, of course) whose density in bands M2 and HF usually is critical. Remember that the previous compression stages are of “hard” processing. Very low frequencies get a “punch” with the “Bass Clipper” control at -4.3 dB

06: ROCK LATINO

This adjustment was thought for Rock and Pop music with strong presence of Indo-afro-Latin-American percussion instruments. By such reason, the attack of the mid band “M2” was freed and, in smaller proportion, the high band (approximately 16 mS and 12 mS respectively). This increases the presence of that bands, in which great part of the spectral components is concentrated that contribute to “definition” and “brightness” to the instruments before mentioned. The effect reinforces with a fastest recovery in the high band (28 mS).

The attack for the low band is also something slow (65 mS) whereas its recovery is of the order of 1/2 second. The objective is to obtain basses with great “height” but without losing the attack, of vital importance in these rhythms, since many times the bass is who takes the cadence of the song.

This adjustment can be considered a “moderate” processing, since the AGC output AGC is fit around 8 dB.

This adjustment is very appropriate also for Son, Salsa and its derivates. For these styles the processing can be increased increasing the output level of the AGC until around 12 dB.

07: ROCK IMPACT

It’s the ROCK program but with a little more processing.

08: POP

This is another “moderate” preset. Designed for POP music, which include very many styles, this adjustment looks for obtaining very good loudness and a “flat curve of EQ”, with good dynamics.

The AGC level is around 7 dB. The density EQ reaches its maximum value in the band M1, with -7.8 dB. As was already said, the equalization curve compensates the response to obtain a cuasi-flat curve.

Respect to the compressors, the low band is freed to obtain great impact in the beats. This is obtained with a slow attack time (49 mS) and a fast recovery (340 mS). In this way, as the recovery of the compressor is fast, each beat of big drum is affected by the attack time, that lets pass the initial impulse of the wave. This impulse that “escapes” is contained soon by other stages of the processor (limiters). But the result is a sensation of greater dynamic range. If the recovery time for the band LF is slow, the attacks of the successive beats would be “squashed” by the compressor (the attack acts only on the first beat and never more, which is equivalent to have an attack time equal to zero).

The high bands also are something released, to obtain “brightness” and presence in trebles, something of extreme importance in POP music (usually strident, cheers and “up”). The “Bass Clipper” threshold is increased up to -3,5 dB.

09: POP IMPACT

This is an adjustment for generic music. The gain of the AGC is 6 dB and the BASS CLIP threshold is -4 dB to reinforcement the lows.

10: CHILL-OUT

This can be considered a “moderate” processing. “Chill-out” makes reference to a calm music, generally of electronic instrumentation.

By the nature of this music, the commitment with the spectral balance is not capital, that is to say, there is more freedom to make the adjustments. In this case was chosen to emphasize a little the lows and the highs presence. This last one is obtained making fast (60 mS) the recovery of the high band, whereas the attack of this band adjusts a little slow (20 mS). This causes that enters in action the fifth band of high frequency post-processing (HF). The “weight” in the basses obtained with a fast attack and a slow recovery in the low band and increasing the gain of this band in the density EQ.

Respect to the AGC, the output level is located about 9 dB, with slow attack and recovery times.

This adjustment also is useful for “new age music”. Although these styles are stylistically different, share the use of synthetic atmospheres and moderate to slow tempo.

11: REGGAE

Is a variation of “Chill-Out” preset with higher “Bass Clipper” threshold (- 4 dB).

12: BOSSA NOVA

This is a moderate preset. Attack times are fast and recovery times are moderate.

13: MAX LOUDNESS

This it is the more radical adjustment. As the name of the program suggests, the primary target was to obtain a great loudness on the air (and improved reach of the FM), leaving in background the “clarity” of the sound and the spectral balance.

The output level of the AGC is located in 15 dB, whereas the density EQ reaches -6.5 dB in the band M1 the highest adjustment between the 15 default programs.

Since the gains of the AGC and density EQ are high; the applied compression is something soft, that is to say, fast attack times and slow recovery times. In this way the signal is more compressed. It is denominated “soft” because when a compressor acts, for example M1 on a guitar, the compressor reacts immediately, to have a fast attack, and maintains with its gain reduction (slow recovery), containing the successive attacks of the guitar, that are not affected by the attack because the compressor still did not recover.

Obviously, the dynamic range perceived in this type of adjustment is lost almost completely, basically due to the compression.

The combination of a hard compression with high gains will cause a heavy processed sound, which will sound rough on the air with certain type of music.

14: VOICE IMPACT

This adjustment specially is designed for locution. When 462dsp is connected to the console using MICstart; opening the microphones 462dsp changes its program to process the voices with a special adjustment.

The **AGC** must recover fast to compensate, for example, a telephone communication that arrives with low level. The hold threshold must be relatively low, around 17 dB, to avoid that the AGC “is hooked” and, in this way, to obtain that it reacts quickly before any difference of level between different voices. Remember that the objective is that the voices always sound on the air with the same level. The attack of the AGC also must be fast, to make level abrupt changes of level that can take place by shouts (discussions) or outbursts of laughter.

The speed of attack of the compressors must be fast, to contain great impulses that take place in the beginnings of phrases, when the compressors begin to work. If the attack times are long, can take place excessive processing and trim of the signal in certain entrances of the speaker; in other words, the speaker will sound “saturated” or “dirty” during a brief moment when he begins to speak.

On the recovery times there is more freedom, reason why they will be fit according to the type of voices of the radio. Like general rule, remembers that long recovery times produce “a smooth” processing, whereas with short times the loudness is increased but the processing becomes more “heavy” (greater compression).

Respect to the gains, **the voices do not tolerate high multiband compression**, so if for music a strong” adjustment is used “, for the voice the AGC output level will have to be located around 9 dB. An excessive processing will sound disagreeable to the ear.

TAKE IN MIND

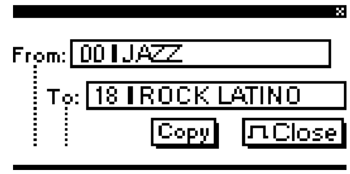
- When creates your own adjustment for the voices, considers that the equalizing does not have to be radically different to the used for music. That is to say, the voice impact program MUST BE CUSTOMIZED maintaining the equalization used for the music.

4.3 USER PROGRAMS

Default programs cannot be modified. To make your customized adjustment, look for that preset that more approaches to the sound that you are looking for your radio; and copy it to a free user memory. Soon modify the preset values in the user memory. To copy a program proceeds as following:

4.3.1 COPY PROGRAM

At the main menu enter to the option COPY PGM. A screen will appear to define:



a) **Source program (From):** Number (name) of the program that you want to copy. To select this option turns the JOG (observes the arrow) and touch it briefly to confirm. Soon turns the JOG again to explore the programs, and touch briefly to confirm.

b) **Destiny program (To):** Number (name) of destiny program. To change it proceed like in the previous step.

In order to confirm the action, turn the JOG to select “Copy” and touch it briefly. The processor remains in the present program. In order to load the copied program you needs select it from the main screen.

Once a user program is overwritten, is not possible to recover it. Default presets cannot be changed by any error of use.

4.3.2 EDIT PROGRAMS

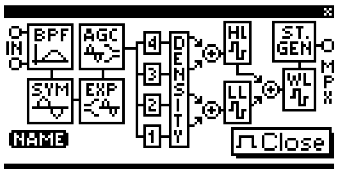
From this option you accede to the heart of the processor. You can edit the processing values for anyone user program. Selecting this option you enter to the program edition screen, described next.

TIPS

- To people who do not have time to fit and to hear carefully the Radio, throughout several days, we recommend to use some of the 15 default presets. Only sound technicians with great patience will have to try to make their own adjustments. Its patience will be compensated surely by a customized sound different from the sound of other radio stations.
- Please read carefully the description of each adjustments made by Solidyne. Take it the time to listen to each program and how each control affects to the sound on the air. Once you knows clearly “so that” of each adjustment, comes to make your customized adjustment. This way will be easy for you to obtain that sound that you have in mind.
- Of no way is recommendable to change values “completely without information”, without knowing how will affect each change to the sound of the radio.
- Finally, an encouraging commentary; these adjustments are zero risk, because you always will be able to return from the default program, that never change.

4.4 PROGRAM ADJUSTMENTS

The screen “Edit Program” shows the blocks diagram of the processor, showing the main stages involved in the audio processing. The operation is the same one that in other screens. Turn the JOG to select a stage and touch briefly to enter.



LISTENING ON THE EDITION

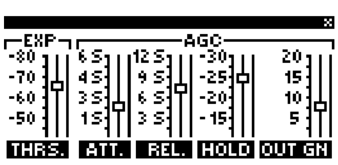
- Next we will see the different forms to create or to modify the sound of a program. Each change that we make will listen on the air. Therefore we will have to listen carefully with a good receiver (better still tuning Walkman or and an audio amplifier of high quality, with good loudspeakers). Not to listen the signal directly from the audio output of 462dsp, but the air signal of the trasmisor

4.4.1 PROGRAM NAME

Allows change the program name. Press briefly on this option to enter to the edit mode.

- The characters are changed turning the JOG. To confirm press briefly the JOG.
- To confirm the changes, press and hold on OK.
- To cancel and leave this screen, press and hold on any character.

4.4.2 EXPANDER / AGC



4.4.2.1 AGC OUT GAIN

The output level of the AGC is a CRITICAL ADJUSTMENT, since it determines the level with the signal reaches the multiband compressors, and therefore the compression degree that will be applied.

So the action of the multiband compressors fits using the output level control of the AGC. LOUDNESS BANDS indicators show the compression degree applied by the multiband system. Each band has its own indicator.

Usual values for the AGC output are between +6 to +12 dB.

TAKE IN MIND

- How much greater the compression degree is, more loudness has the program on the air. But high compression levels (greater than 18 dB) can produce a “confused” sound in certain type of music (hyper-processed, of high density) and excessive alteration of the timbre (with soloists’ voices or instruments). For this reason it's a critical adjustment and you must pay special attention to this control.

4.4.2.2 AGC attack time

The *attack time* is the time that the AGC takes to reduce its gain when the input signal increases. Like general rule, it can say that for voices short times of attack must be used (500 ms); whereas for music longer times are preferred (3 to 4 seconds).

Note that when the input signal increases quickly, during the AGC attack time the signal is contained by the multiband compressor that will act strongly until the AGC compensates their level. Depending on the adjustments of the following stages, a too slow attack of the AGC can cause an excessive compression of the signal (mainly with voices)

4.4.2.3 AGC recovery time

When the input signal decreases, the AGC begins to increase their gain to compensate the fall of level at the input. Remember that the objective of the AGC is to assure that the signal reach the processing stages with a very stable level; independent from the console's level. The time that takes the AGC in compensating the gain reduction is called recovery time.

TIPS
• In order to process voices, use slow recovery time, so that the AGC can effectively correct differences in the program signal. Let us analyze an example: the telephone line arrives with low level. While the speaker in studies speaks, the AGC works at certain level; when the caller speaks, the telephone presents less volume on the air, and the AGC must act fast to correct this situation; increasing its gain. When the speaker returns, the AGC will reduce their gain again. And it will have to act with very little delay. The hold threshold will have to be low, to avoid that the AGC "is hooked" freezing their value during the telephone communication of previous example. • The AGC attack and recovery times must be carefully defined so that its action is not in evidence. If the attack time is excessively slow, the action of the AGC could notice (the level reduction can notice). If the recovery time is very slow and the attack time is very short, when somebody shouts (a cough, an outburst of laughter) the AGC reduces its level abruptly and takes soon in recovering its level. Then the effect will be similar to "somebody lowered the volume of the radio". • For music, it agrees that the recovery time be slow. If it's fast, the dynamic of the music are completely lost.

4.4.2.4 Hold

This is a gated AGC. For this reason, when the input signal falls abruptly, the AGC does not compensate its gain, but that congeals their current value; remaining in that state until the signal exceeds the "hold" threshold. Otherwise the AGC would compensate the input level continuously, increasing the background noise in the pauses; due to the signal absence the AGC would increase its possible gain to the maximum. The Gated AGC solves this disadvantage.

Strictly speaking, AGC value is not congealed; but that slips slowly towards 0 dB; to avoid that it is hooked if the signal remains with low level (slope = 0.75 dB each 13 seconds; 4.5 minutes from -15 dB).

On the other hand, you can adjust the HOLD threshold to conserve part of the dynamic range of the music. That is to say: if a "forte" passage is followed by a subtle appearance of an instrument, the AGC will hold its gain level, giving rise to the contrast of loudness. When the "piano" passage reaches the HOLD level, the AGC unhold and begins to increase their gain according to the recovery time.

4.4.2.5 Expander threshold

Change the expander threshold; from -50 dB to -80 dB in steps of 1 dB. This level is referred to 0VU input (overload level). The threshold is the point from which the expander begins to reduce its gain, as signal level is reduced. The object of the expander is to improve the signal/noise relation on the air. This is because the multiband compression, although increases the loudness, reduces relation S/R. This effect would be annoying, but for the action of the expander.

4.4.3 MULTIBAND COMPRESSOR

The object of the multiband compression is to increase the energy in the entire audible spectrum. The theoretical foundations of this technique are explained in Chapter 6 of the 462dsp user's manual. Next a brief review of multiband compression offers, and is detailed soon how different processing adjustments affect to music and the word.

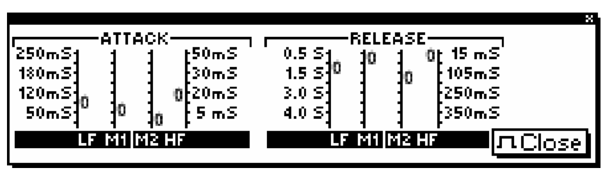
Remember: a compressor is basically an amplifier whose gain changes from a certain level of signal. This change of gain denominates "compression ratio". For example, a compressor with a 2:1 ratio will produce an increase of 1dB at on the output when an increase of 2dB takes place in the input. In this case the increase of signal on the output was half with respect to the increase on the input. The level from which the compressor begins to be not-linear calls "compression threshold". When the signal surpasses the threshold, the compressor begins to work..

On the other hand, when the signal surpasses the threshold, the compressor takes a time in "reacting" to modify its gain. This time is denominated "attack time"; and during the attack the relation input/output pass to be linear to work according to the compression ratio.

When the input signal reduces, the compressor takes in recovering its original gain, that is to say, during a short time it continues compressing. This time is called "recovery time".

In 462dsp the compression thresholds are the same for all the bands, to not alter the sonorous balance. The compression degree simultaneously adjusts for all bands, changing the level with which the signal enters to the compressors (AGC Output). As greater is the level, greater is the compression. You can set the attack and recovery times for each band.

Have in mind that the settings for the compressors and density EQ's varies according to the type of program material, reason why there isn't a unique adjustment 100% optimal for all musical styles. A great feature allows remote control from the automation PC (using software Solidyne Audicom7) to change the processing program according to the music style. Remember that VirtualRack 5 also allows defining program changes according to a scheduling. Next, a screenshot of the multiband compressor stage and soon its description:



4.4.3.1 Attack times

It's the time that takes the compressor in acting, once the signal overpasses the threshold. Slows attack times gives more “impact” to the sound, but greater will be the action of the limiters. This is because the attack of the sound passes through compressors and arrives at the limiters with high level. The limiters contain the peaks by clipping, a technique much more lasts that the compression. With short attack times hard clipping is avoided, but very short times can produce a sound too ´flat´ for certain musical styles.

The scales for attack times are different for each band.

In order to define the attack times, you must consider the type of material to process. Some musical styles, like the rock & pop, tolerate a hard processing (more clipping). This offers a great sensation of dynamic range (depth of the sound). For orchestral music, jazz, piano, agree to use faster attack times

4.4.3.2 Recovery times

It’s the time that the compressor takes in recovering its linearity as soon as the signal falls below the threshold.

The recovery times also are key settings, to produce the perception of dynamic range. In main lines; if the recovery is slow the compressor practically works continuously; and initial impulses produced by the attack time are lost. That is to say; the attack takes place the first time, but as the compressor's gain does not recover, the following attacks are fully compressed. In this condition the attack time does not have any effect. The attack time is responsible to generate “sensation of dynamic range”.

In percussion this is not desirable. For example: techno music requires short time of recovery time in the low band, not to lose the attack of groove. The compressor must recover so that each "beat" is affected by the attack time.

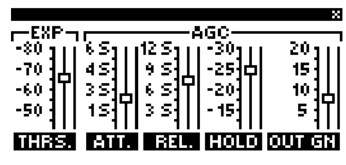
In the highs band the same occurs. If the recovery time is very slow, the attack does not take effect and the loudness of the high band (a Hi-Hat, for example) diminishes. Faster recovery times increase the loudness of the highs and the “brightness” of the sound, but it can be a little rough sound for some musical styles.

Note: The attack, like the recovery, has a scale for band LF and M1 and another one for bands M2 and HF. Band SF copies the scale from HF.

4.4.4 DENSITY EQUALIZER

This technique eliminates the problem of the conventional console's equalizers, whose action soon is canceled by the multiband processor (since it compresses what the equalizer reinforces). The technique of Density EQ operates in combination with the multiband compressor. Therefore its adjustments are interactive.

Density EQ **works creating four floating thresholds for the average sonorous energy**, in four frequency bands: 80 Hertz, 1 KHz, 6 KHz and 12 KHz. This equalization allows to obtain, still in conditions of high loudness, a contour of equalization based on the energy of bands (instead of differences of relative gain between the bands like in a conventional EQ).



To change a value, select a band and touch briefly the JOG (note that the arrow becomes a pencil). Turn the JOG to move the slider.

The **dotted area** is only for laboratory testing. It is not a safe operating zone for audio processing.

Bass Clipper (Punch)

This control changes the threshold of a low band limiter. The change of this threshold allows to brings “punch” to the music’s below 80 Hz. Increasing this threshold you increases the bass peaks, but not the general bass loudness in a noticeable way. You must use a monitoring system with good low response to hear this adjustment.

Values over -4dB must be carefully tested according to the music that your radio station plays, so it can produce audible distortion with some music’s styles. All default presets uses this control at the minimum (– 6dB) except “Melodic Hi Boost”; “POP”; “Night FM” and “Rock”.

REMEMBER
• Changes occurs in real time, that is to say, as you modify the value, the change will be listened on the air.

Chapter 5 Remote Control

462dsp can be controlled from a PC, by means of RS-232 connection. The control software "VirtualRack 462dsp" offers numerous features:

- To change the current program to anyone of the 30 presets from the PC using a drop-down menu. The change take place in real-time.
- Create and edit the programs with an advanced graphical interface.
- Save the factory presets and user programs to the HD, as backup or to transfer them to others 562's units.
- To automate changes of processing according to a scheduling, to use different processing programs from different hours from the day. You can increase the loudness of your radio in the rush hours of car traffic to arrive more force at the radio receivers from car when they are in the centric zone. For other schedules smoother adjustments can be used, and finally last in the night to set a very smooth programming, to accompany music. The frequent change of the processing ways tends to fight the auditory fatigue that happens when the radio sounds always equal throughout the day. **Note:** This feature requires a PC connected to the 462's, running 462dsp VirtualRack..
- Manage the 462dsp processing programs from the Solidyne Audicom7 on-air automation software. You can change the programs according to the musical style.

The software runs on Windows 2000/XP. For more details about this software, please refer to the on line help contained into the CDROM.

5.2 Set up and connections

Interconnection is made using a standard cable RS-232, with DB-9 female connectors in an end and male in the other. The computer must have an available serial port.

VirtualRack software automatically recognize to the 462dsp, by exploring all RS-232 ports.

An **USB to RS-232** adaptor of good quality can be used. It is obtained in any computer's store.

Please refers to the on-line help for details about the software.

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NOTE: *in order to complement the study of this subject is recommended to visit our WEB (www.solidynepro.com). In the DEMO section there are a Power Point presentation called Audio Processors. It has a complete Technical Appendix that it analyzes how the audio processing increases the coverage area of the FM stereo transmission. Also we recommended to read in our WEB the article "Theory of the multiband processing"*

2.1 A brief history...

From mid of the 1930 decade, when appears the first compressors and expanders units, to the present time, all chains of audio for broadcasting incorporate devices whose function is to alter the dynamic range of the sound. The advance of the technology improves these devices during the '70s.

The compressors, expanders and audio limiters were gaining in efficiency and complexity. In the beginning, its main parameters (attack and recovery times, thresholds, etc.) were fixed by design or by the operator, through the device's controls. In the '70s, these functions begin to be automatic, based on the characteristics of the audio signal, but having at the same time a control on their action to be able to customize the sound.

When five or more devices are grouped in same equipment, they begin to be denominated: AUDIO PROCESSORS.

Since 1970, **Solidyne** introduces important advances in this field, like the invention of a control technique based on FET's with guided gate (see publication in Rev. Tel. Electrónica, September/70). They follow diverse publications, having particular international relevance the work published in June/76 at the *Journal of the Audio Engineering Society*, New York, U.S.A. where a new concept was introduced, that persist to the present time: **PHSICOACUSTIC PROCESSING**.

This new technique is the base for all the modern audio processors for broadcasting use. The necessity to process the phase to make symmetrical the human voice waveform is another one of the techniques that Solidyne has introduced internationally (see mentioned article AES). Today, our ideas are used by Orban, Omnia, Aphex, etc.

The concept of psychoacoustic processing is simple in essence, although of complex accomplishment. It consists of analyzing the way in which the sound is perceived by our ear, considering diverse investigations and developed acoustic models.

The brain uses to process audio data, the information that arrives through 30,000 nervous fibers, originating at the Basilar membrane. Then, it will be possible to be computed the auditory reactions and to be governed all the aspects of the audio processing. This way, the electronic system works transforming the original signal into another one, of greater energy and greater quality of sound. Then, it will be possible to reduce the dynamic range of the audio signals, to eliminate the peaks, and even, to clip them partially to increase its energy.

If this were made directly, obeying to purely electronic concepts of efficiency, the quality would be degraded and the sound would be very poor. If, however, the psycho acoustic concepts are applied, and factors like the aural masking, the pre and post pulse inhibitions, the Hass effect, the reflections at the ear pinna, the aural models of Dr. Karjalainen, etc; it will be possible to create a new generation of processors that allow to important increases of energy, increasing at the same time the sensation of "Perceived Sound Quality".

At the light of these discoveries the psychoacoustics processing was defined in these terms:

PHSYCO ACOÚSTIC PROCESSING is the technique that allows to increase the range of AM or stereo FM transmission, by increasing the energy of the audio signal, and also increasing the "quality of sound" perceived by the listener.

Nevertheless, it is fundamental throughout this process, to maintain very low the audio distortion produced by harmonic and IM components. This happens because the psychoacoustic processing MODIFIES the waveform of the complex signal of audio, but IT DOES NOT DISTORT IT. Since the distortion concept, in this context, implies the existence of a sound that offends the ear, sounding unnatural.

This is because the psychoacoustic processing obtains that the ear accepts like of better quality than the original one, to certain modifications of

the waveform. But it does not mean an "anesthesia" to the ear, to avoid perceiving the distortions due to deficiencies in the quality of the electronic circuits of the processors.

Considering that to obtain an excellent processing is necessary, at the moment, to use between 7 and 10 stages of processors, the distortion of each stage must be smaller than

0.01%. Greater distortion values, will lead inexorably to a degradation of the sound quality. You must remember that has been demonstrated (Journal of AES, Vol. 29,4,p.243), that is possible to measure distortions of 0.05% through a common loudspeaker (distortion bigger than 3%). This demonstrates that one distortion do not mask another one. A practical rule is, then:

ALL DISTORTION INTRODUCED IN THE AUDIO CHAIN OF THE TRANSMITTER, THAT EXCEEDS 0.05%, COULD BE LISTENED BY THE AUDIENCE, EVEN THROUGH RECEIVERS THAT HAVE VALUES OF DISTORSION 50 TIMES GREATER.

This, of course, is not a novelty for the conscious audio engineers around the World. Therefore, the line of SOLIDYNE processors has distortion values smaller than 0,02%.

Friendly interface

462dsp has many of its automatically fit functions, under control of the program of audio. But there are essential controls for "ing personalize ng personalize" the sound of its radius, that are fit by the user.

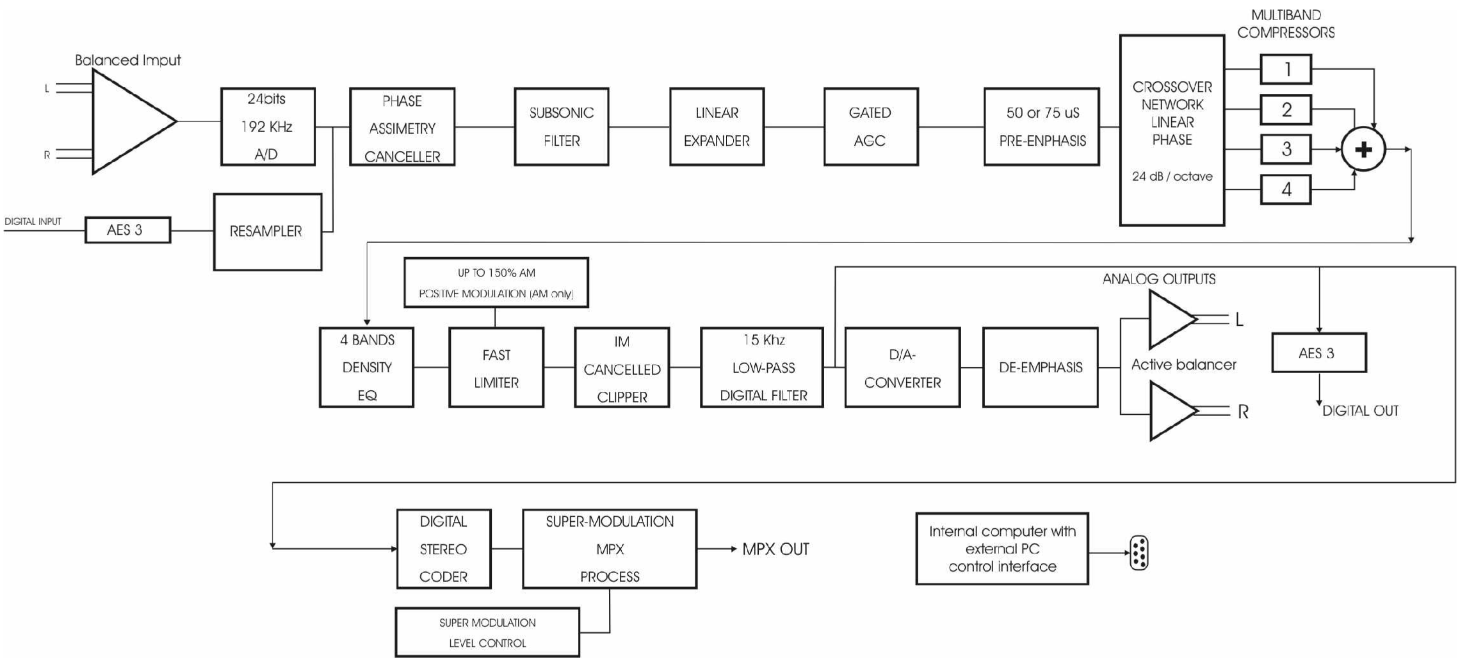
The unit has several programs fit in laboratory for an optimal performance with different types from sonorous material. These programs allow you to be on the air in immediate form.

Soon, you can copy the program chosen to a user memory, and modify the adjustments to customize the sound.

When finalizing the adjustment, assigns a password to the equipment to maintain the configuration unalterable. Only the engineer who has the password can alter the sound of the radio.

2.2 Solidyne 462dsp psycho acoustic digital processor

2.2.1 Blocks diagram



2.2. 24 BITS @ 192 KHZ A/D CONVERTER

This module converts the analogical audio signal to digital audio, with 24 bits of resolution with a sample rate of 192 KHz.

Digital Input / Resampler

AES-3 input supports resolutions of 16/24 bits and Fs from 32 up to 208 KHz.

An stage called *resampler* converts the digital input signal to 24 bits / 96 KHz; the resolution that the processor uses internally.

From here and until the signal returns to the analogical domain (D/A converter), all process descript are made into the DSP's using complex algorithms.

REMEMBER
• In case of failure in the digital signal, the unit automatically changes to the analogical inputs.

2.2.3 PEAK SIMETRIZER

It is known that, by a particularity disposition of the vocal cords, the sonorous emissions that these generate are asymmetric triangular pulses. The three cavities that filter and shape these formants, to obtain the vocal sounds, do not modify this intrinsic characteristic of the human voice. All the spoken word and still sung is strongly asymmetric.

This creates an important reduction of the energy of the audio signal, particularly when it pass through a compressor, because the compressor sets its compression level for the greater peak, does not concern its polarity. In this way, when a polarity is fit to the 100%, the opposite polarity hardly surpasses 50%, due to the asymmetry. The fact that the music sounds louder than the human voice, after pass through a compressor is a phenomenon well-known. This is because the musical sounds are symmetrical, whereas the human voice is not.

In order to correct this problem, WITHOUT INTRODUCING ANY ALTERATION AT THE SOUND QUALITY, peak asymmetry canceller is used.

This technique, based in a discovery of the Dr Leonard Kahn, acquires international validity with the work of Oscar Bonello, published at the Journal of AES, Vol.24,5 in which it is described, for the first time, the theory of its operation.

The peak asymmetry canceller is in essence an all-pass network, a class of not minimum phase network. That is: a network whose transference function has zeros in the right semi plane. This network has a full flat response to frequency; only its phase response is function of the frequency. This phase rotation, which must compliment a very particular condition, is responsible of the peak symmetry of the audio signals. Those signals that by their nature, are totally symmetrical (like most of the musical instruments), are not modified by this processor. This processor, by itself, allows to increase between 3 and 5 dB the final power broadcast by your transmitter (it is to say that it increases by a factor of TWO the average power transmitted). Numerous tests have been made in different countries, to verify, in real conditions, these results.

2.2.3 SUBSONIC PHILTER

Is a Chebyshev type high-pass filter, with low ripple. Its cut-off frequency, of 20 Hz, eliminates the signals of audio below that frequency. These subsonic components do not contribute anything to music, since they aren't listened by the ear. Nevertheless, they have a pernicious effect that produces a disagreeable sensation: the saturation of the amplifiers and the loudspeakers (by excessive excursion of the cone). The sound that is obtained with the connected filter is pure and clean.

2.2.4 LINEAR EXPANDER

The expansion, previous to the compression process, is an excellent resource to increase the signal/noise ratio of the original program. This is advisable, since the compression process, when reducing the high level passages, consequently increases the relative level of the passages of low level, and therefore the noise. This is a forced consequence of the compression process that has particular effect in the increasing of the ambient noise of the microphones. In order to avoid that, the Solidyne processors incorporate a linear expander, previous to the compressor stage.

The expander works within a very wide range of signals, below a threshold value. That means it always expands within that range, for any level of signal. The curve of transference, based on the input level, is a straight line (from there the "linear" name). While the signal stays below the threshold; by each 10 dB that the input level reduces, the expander will reduce, for example, 3 dB additional (1,3:1 ratio), that is to say that the output will be reduced in 13 dB. This happens for any input value, below the threshold. Then if the input is reduced in 30 dB, the output will do it in 39 dB; that is to say that the background noise has been reduced in 9 dB. This way, the expander compensates the increase of the background noise that the compressor, like undesired effect, will increase.

At this point, maybe you will be thinking that DOES NOT HAVE SENSE to make an expander of the signal and soon to compress it. You will think, perhaps, that an effect cancels to the other. But it's not true for two reasons. First: the different attack and recovery times. Second: multiband compressors have elevated threshold, whereas the linear expander has a very low threshold and a linear behavior below the threshold. It means that the actions do not cancel, because both processes are not complementary.

The linear expander, to optimize its behavior, has instantaneous attack and a fast recovery times. Here is where the psychoacoustic concept "post-pulse hearing inhibition" is used. This allows using an expander with a quick recovery time, so that it's not perceived by the ear. The broadband compressor that follows the expander has a very slow recovery time. Therefore, with impulsive signals, as the audio program, any cancellation effect occurs.

Another advantage of using a linear expander previous to the processing is that an excellent audible sensation of dynamic range is obtained. In fact, recent studies have demonstrated that the audible sensation produced by the level variations of an audio signal, is related to the changes happened in the first 50 milliseconds, and is little dependent of the reached final value. This implies that an expander in the short term is perceived like a great dynamic range, whereas the power sensation (and even the coverage area of the radio transmitter) is related to the AVERAGE ENERGY, which depends of the compression of the energy level.

You can see that they are two concepts different. With audio processors of conventional design, the expander and the compression were antagonistic concepts. This does not happen in the field of the psychoacoustics processors.

2.2.5 GATED AGC

This stage constitutes a digital compressor AGC type (Automatic Gain Control) with its constants (slow) based on the program signal. This adjustment is made having in mind the psychoacoustics criteria before explained. This is very important since the AGC is a wide band compressor; the ear tends to note its action due to the modulation that bass frequencies produce in the highs. In order to avoid it, the AGC time constants are carefully controlled. Still more; we cannot speak of attack or recovery times in strict form, but a *curve of control,* since the way of the attack or recovery ramp is controlled according to these criteria

Otra característica importante de este compresor es que su recuperación es GATILLADA. Es decir que solamente acciona en presencia de señal de programa. Cuando el nivel de programa cae por debajo de cierto umbral con respecto al nivel de salida del AGC, la recuperación es cancelada y el compresor permanece exactamente con la misma ganancia que tenía antes de caer la señal. Esta característica permite un funcionamiento más natural del procesador, puesto que evita que en las pausas entre los diálogos, por ejemplo, se note la 'respiración' del AGC (incremento en el ruido o soplido de fondo). O que al terminar un tema musical, se escuche un aumento del ruido de fondo antes de comenzar el tema musical siguiente, etc.

2.2.6 MULTIBAND COMPRESSOR

The purpose of the multiband compressors is to increase the perceived loudness sensation. The human voice and music will sound more solid, with better dynamic balance. Still more, the increase of the average energy of the audio signal is very considerable, increasing the coverage area of the radio for A.M. and FM transmissions (for more info please visit www.solidynepro.com).

Multiband technology bases on the studies of Stevens (ref 1.2.3) about the loudness of each band frequency and the studies of Zwicker (ref 4) About its relation with the Critics Bands of the human ear. The integration time of the ear to reach the maximum loudness is of the order of 200 milliseconds (ref 5). This time must carefully be incorporated to the controls of the loudness compressors, to obtain the desired effect. The ear will perceive a greater loudness when the band compressors increase the relative loudness level.

The processor Solidyne 462dsp has digital frequency splitters with Butterworth filters of 24 dB/octave that divides the program signal in five frequency bands: lows; low-middle; high-middle; highs and super-highs. The sub bands high-middle and highs works in combination with the 5th band super-highs, processing the highest frequencies of the audio spectrum. This 5th band is of great importance since take incidence over the peaks of the signal, one of the most critical processes carried out by any audio processor.

The controlling of the dynamic range using band frequencies offers great advantages:

1. To increase the total energy, by the use of fast compressors for bass and ultra-fast for treble. If the bands were not divided, the compressors with so fast recovery time would produce a disagreeable sound effect; the percussion of low frequencies would modulate the high notes. And the high notes of an instrument would as well modulate the low tones, of a violoncello, for example. The increase of the energy increases the cover area, that is to say, the reach of a stereo FM station. See demonstration in paper of O. Bonello, AES Journal, New York, March 2007

2. Allows to increase the perceived loudness. This is because most of the modulation capacity of a transmitter or audio amplifier in general, playing Pop music, is generally used by low frequencies signals, below 160 Hz. Nevertheless this information contributes very little to the loudness sensation, due to the reduced sensitivity of the ear for those frequencies. Therefore is desirable to increase the level of the mids and highs frequencies. But this cannot be obtained by simple equalizing, since the sound balance would be destroyed. The compression in separated bands allows to increase between 6 and 12 dB the energy for high frequencies without altering the tone balance. In fact, the frequency response continues being flat.

3. Processing completely eliminate the "flat sound" sensation, perceived when a sonorous material is compressed, by means of fast compressors. This is obtained, additionally to the division in bands, using attack times appreciably elevated. This allows that very short peaks of the audio signal arrive freely to the following stage (peak limiter) which eliminates them, but maintaining the

psychoacoustic sensation of power associated with the audio peaks. This is related to the *Burst Masking* effect; recently discovered (Ref. 8).

2.2.6 DENSITY ECUALIZER

This technology operates in 5 bands modifying the density of energy (instead the level) of each band. Is formed by complementary filters of 24 dB/octave carefully designed to obtain a response variation smaller than 0,2 dB. This built-in equalizer offers an enormous flexibility.

By example: in the case of FM transmission; is well-known that the use of EQ at the console output has an adverse effect in the sound quality, since the more a frequency band is emphasized, grater is the action of the audio compressor (previous to the transmitter) for that band. Equalize a band implies to unbalance the entire audio spectrum. It doesn't happen with DENSITY EQ, since its action is coordinated with the following stages. The boosting of a frequency band is translated then in a correlative modification of the multiband compressor threshold, to carry out the new equalization. In this form, its action extends to the range of sounds of very high intensity, where the conventional EQ's are inefficient, due the excessive compression.

2.2.7 IM CANCELLED CLIPPER

The analogic era...

One of the techniques most well-known and used to increase the energy of an audio signal is the use of an audio clipper. With its evolution, the analogical processors was optimizing this technique to reduce the udesired effects caused by the clipping. The problem appears because when an audio signal is clipped, it generates a high number of harmonic components and intermodulation (IM).

The psychoacoustic studies have demonstrated a high tolerance of the ear towards the harmonic components, increased by the aural masking that the multiband processor allows to obtain. In fact, all the musical instruments, as much as the synthesized ones, have a high amount of harmonics, superposed to their fundamental notes. Therefore adding harmonics artificially, increases the wealth of the musical timbre.

Opposed is the effect that the distortion by IM produces. This type of distortion DOES NOT EXIST in the musical instruments, and its effect is highly disagreeable and irritating for the ear of the audience. For the exposed reasons, considerable efforts in research have been made to obtain audio clippers in which the intermodulation is reduced or even better, cancelled, of being possible. System RIMCA, developed at the laboratories of Solidyne, begins in a multiband clipper, was a good solution used in analogical units.

The digital days...

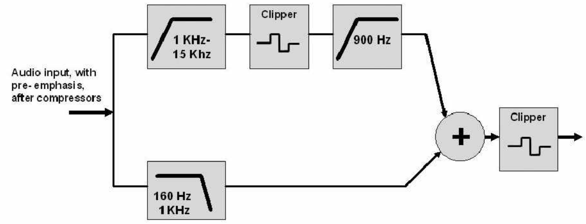
In digital audio, the adverse effects derivates of the clipping, far from disappearing, becomes more complexes.

DSP technology, since works with digital signals, has limitations in to manage nonlinear processes (as it is the case of the clipping, that it generates signals with very steep slopes). These type of processes generates harmonic components of very high frequency that violate the condition of Nyquist, degrading remarkably the final audio quality. The conventional digital clipping techniques, used by other marks, work resampling the signal at greater sample rates, thus to be able to reduce (but to never eliminate) to the distortion caused by the alias effect.

Although the result that is reached with these techniques is acceptable, the "collateral effects" caused by digital clipping are diminished, but not eradicated completely. For this reason, the laboratories of Solidyne continued developing new techniques.

After long investigation and numerous tests; 462dsp took a step more in the field of digital clippers; using a new system of 4 channels with 10 MHz of bandwidth (that is equivalent to 32 bits with 20 Megasamples), that obtains that absolutely clean and crystalline sound; free of spurious and stridencies caused by aliasing.

For it we used a configuration like the FIG-4 in which we have two cut channels. The channel of serious and average, on the one hand, does not cause intermodulation. The channel of high frequencies, main person in charge of the IM, has a filter that eliminates all IM below 900 Hertz



2.2.8 LOW PASS 15 KHz DIGITAL FILTER

In order to be in agreement with the standards for FM stereo transmission, a 15KHz low-pass filter is at the input of the stereo coder, which produces an attenuation > 50 dB at 16.6 KHz; to avoid interference with the RDS sub-carrier; and > 60 at the pilot tone (19 KHz).

Is an digital elliptical filter; of great stability. This filter also guarantees that not exist spurious components at the RDS band of 57 KHz.

2.2.9 DIGITAL STEREO CODER

It uses digital technology to generate the MPX signal. This technique, created by Solidyne, allows obtaining a coder with excellent features; with distortion 10 times below the audibility threshold and channel separation better than 75 dB.

It's based on the *oversampling* concept, that divides the audio signal in 16 samples that are processed separately at 38 x 16 = 608 KHz. Due to this elevated sampling rate, the anti-alias filters works over 500 KHz, eliminating the phase rotation effect that take place at 53 KHz, which no allow to achieve a good channel separation. With this new solution and the use of advanced technology in each part of the circuit, residual components of distortion below -90 dB are obtained.

It is described separately in this manual, the way to do measurements and reception tests of the stereo coder (see Chapter 7).

2.2.10 MPX PROCESSING

The studies about the modulation on an FM transmitter indicate that when the transmitter is modulated by stereo MPX signal appears a new effect, not present on the original audio signal. This effect, called MPX Interleaving (also known as peak correlation), determines that the modulation peak in MPX does not coincide with the modulation peak of the stereo signal, considered in independent form.

Solidyne processors uses a MPX Processing technology named Super Modulation. This processing consists on a system that controls the peaks, operating at 608 Khz, eliminating the peaks in MPX base band signal and filtering them so that there are not left residual components in the audio band.

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6.- Oscar Bonello **.** NEW IMPROVEMENTS IN AUDIO SIGNAL PROCESSING Journal of the Audio Engineering Society, Vol. 24 Nº 5. USA, 1976

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7.1 MEASURREMENT PROTOCOL

Next instructions offer to make diverse measurements in case that it is required to state the engineering specifications of the equipment.

7.1.1 Checking the INPUT LEVELS

Input: Audio generator at 1 KHZ on both input channels (the amplitude is varied throughout the test).

Output: Oscilloscope on XLR audio output.

Procedure:

a) Select the program Nº 28.

b) Adjust the input level of the processor at +8dBm and check that for this input level both input VUmeters be exactly a 1 pixel from the maximum.

c) Check that increasing the input level until +10dbm the VUmeters at full scale. At this value the warning “OVL” must appear under the Vumeters. In the oscilloscope you must observe a perfect sinus wave.

d) Increase the input signal in step of 1dB until +18dbm. At this value the signal at the output must be seen distorted, clipped by the input stage.

e) Check that previous distortion happens just at +18dbm at the other channel.

7.1.2 Checking the OUTPUT LEVELS

Input: Audio generator at 1KHz on both input channels, adjust the **AGC** so it compress 15db.

Output: Connect a dB meter on left audio output (XLR).

Procedure:

a) Select the program Nº 26.

b) Under this conditions, the audio output Hill must be +4dbm.

c) Repeat for the other channel.

7.1.3 FREQUENCY RESPONSE

Input: Audio generator at 1 KHz on both input channels the amplitude is varied throughout the test).

Output: Connect a dB meter to the left audio output (XLR).

Procedure:

a) Select the el program Nº 27.

b) Enter with a sine wave and adjust its amplitude so that the AGC compress exactly 30 dB; waiting around 10 second to the AGC stabilized.

c) Disconnect abruptly the source of signal (HOLD Hill appears on screen) Verify throughout this test that this legend do not disappear at any moment.

d) Make a frequency sweep with an amplitude 25 dB under the final amplitude used in the point b). Under these conditions the HOLD indication must be always on screen.

Realizar el barrido de respuesta frecuencia con una amplitud de 25dB por debajo de la amplitud final utilizada en el punto b. Bajo estas circunstancias la leyenda HOLD deberá permanecer siempre encendida. Whereas no of the 4 bands must enter compression (does matter the input frequency of the oscillator).

e) At the audio output of the processor you must verify a response from 20 to 15.000 Hz with a variation of +/- 0.7 dB.

f) Repeat for the other channel.

7.1.4 S/N

Input: Audio generator at 1KHz on both input channels, adjust the **AGC** so it compress 15db.

Output: Connect a dB meter on left audio output (XLR).

Procedure:

a) Select the program Nº 26.

b) Take this output level as reference of maximum output level.

c) Quit the oscillator and measure the residual noise on the output.

d) You must obtain as least a S/N of 90 dB at the audio output.

e) Repeat for the other channel.

7.1.5 STEREO CROSTALK

Input: Audio generator at 1KHz on left input channels, adjust the **AGC** so it compress 15db.

Output: Connect a dB meter on right audio output (XLR).

Procedure:

a) Select the program Nº 26.

b) Take the output level of the left channel as reference of maximum level at the audio output.

c) At the right output you must obtain a crosstalk bigger that 72dB (difference between the level measured at the point b and the reached in the point c).

d) Repeat for the other channel.

7.1.6 DISTORTION

Input: Audio generator at 1KHz on both input channels, adjust the **AGC** so it compress 15db.

Output: Connect a dB meter on right audio output (XLR).

Procedure:

a) Select the program Nº 28.

b) Under this conditions the audio output must be -2dBm (6 dB below the nominal level of +4 dBm). By the other hand, no of the 4 bands must enter compression.

c) Proceed to measure the total harmonic distortion (THD) It must be around 0.01 %

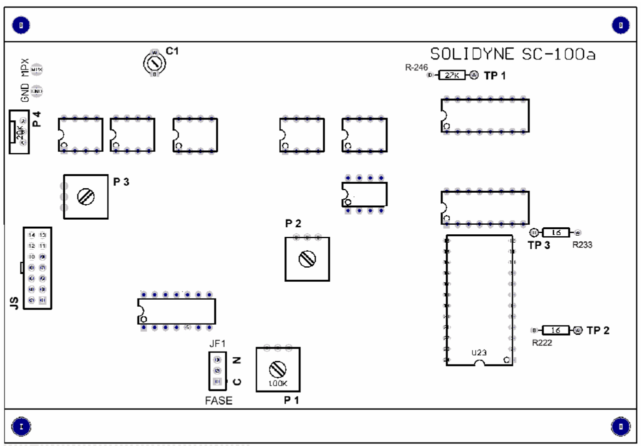
d) Repeat for the other channel.

7.3 SC100 STEREO CODER
Fine calibration protocol

Normally the stereo coder don't need calibration because it's a full stable (after many years) design based on digital oversampling 16x technology. If you wish to make a control or fine tuning, please follow the following steps.

PLEASE! IF YOU DON'T HAVE A GOOD KNOWLEDGE OF THE MATRIX STEREO THEORY OR DON'T HAVE THE APPROPRIATE MEASUREMENT SET, DO NOT INTEND TO CALIBRATE THE STEREO CODER.

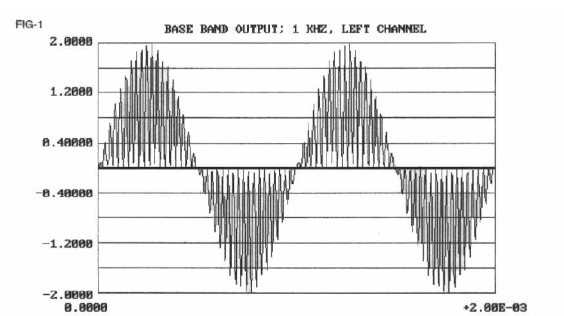
7.2.1 TEST POINTS DIAGRAM



components side

7.2.2 ADJUST PILOT LEVEL

a) Connect to ground the test point **TP1** (R246) in order to eliminate the 19 KHz pilot tone. Use a good DC-20 Mhz calibrated oscilloscope at the MPX output connector. Using the display control, select the MPX output to 4 V peak to peak. Connect a sine wave generator at the LEFT input of the audio processor, at 1 Khz, + 4 dBm output. Verify that the output waveform looks like FIG-1.



a) Replace the audio generator at the processors Left input for audio program (music or voice). Please carefully take note of this peak to peak value; let name it Xpp.

b) Disconnect the audio program from the LEFT input and disconnect the ground of the test point TP1. Then, the 19 Khz pilot tone will appear.

c) With the oscilloscope measure the peak to peak value of the 19 Khz pilot tone; let name it Ppp. The percentage level of pilot tone will be:

Pilot Level [%] = 100.Ppp/Xpp

d) Using the control P4, adjust Ppp to the desired value (normally 9 to 11 % is a good value)

Note: If you have a **Solidyne VA16** modulation monitor or BELAR FMS-2 measurement set, you will be able to measure pilot tone level from on-air transmission and you can correct it using P4.

7.2.3 PILOT PHASE ADJUSTMENT

This method is an *absolute system* not dependent from the calibration of the Modulation Monitor. You will need a very good DC-20 Mhz oscilloscope (Tektronix preferred) with vertical 10x undistorted span. Proceed:

a) Change the **SC-100 PHASE jumper** (JF1) from the Normal "N" position to calibrate "C" position.

b) Input a sine wave generator at 1 KHz, +4 dBm, at Left Channel of the audio processor. Adjust the level of wave generator to get 4 Vpp MPX output.

c) Connect the oscilloscope to the **MPX output** in mode DC.

Use the sine wave output of audio generator, to synchronize the oscilloscope sweep. Adjust the sweep to 5 uS/Div. Modify with trigger level, the center of the wave in order to have it centered at screen.

Adjust vertical sensitivity to 50 mV/Div. You must get a signal like FIG-2.

If not, you must do a slight change in the frequency knob of the audio generator in order to avoid integral multiplication factors that do not allow you to appreciate the correct image.

Calibrate the P3 Phase Control to get zero phase error. The phase error is indicated by the tilt of the imaginary line between H1-H2 points. When this line is full horizontal, the phase is correct.

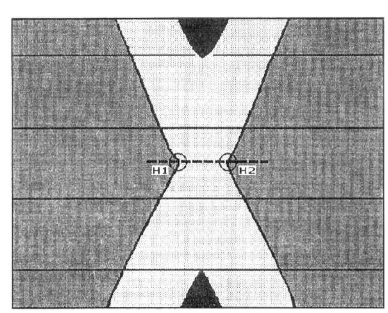


FIG.2 Pilot Phase 10X

e) Change the phase calibration jumper to NORMAL mode ("N" position).

7.2.4 CHANNEL SEPARATION

Due to the high channel separation of the Solidyne SC-100, the only way to recalibrate to original specifications is to use as reference a good measurement set like the BELAR (USA) model FMS-2.

If you do not have an excellent measurement set, please do not intend this calibration. Anyway the Factory Calibration of our stereo coder, do not change with the years.

Note 1: Previous to this adjustment, the Pilot Phase (item # 2) must be adjusted.

Note 2: This calibration must be done with stereo coder inputs disconnected from audio processor. Then, it must be necessary to install a new flat cable connector set, which avoids connection with audio processor and allows direct connection of the audio generator to **JS1** and **JS3** (JS connector).

a) Verify that the input stage bias is correct. Use a DC millivoltimeter to check that voltage difference between TP2 (R222) and TP3 (R233) is less than 0,1 mV. If not, correct it using P1.

b) Connect the BELAR measurement set to the MPX output connector.

c) Check the PHASE mode of the Belar. You must get at least – 65 dB rejection at 19 KHz pilot tone. If not, please make the Belar calibration procedure of PHASE (see Belar manual).

d) Connect a sine wave generator at 400 Hz, 1V RMS to the Left Input (pin **JS1**). Leave disconnected the right input **S3**.

e) Measure channel separation at the Belar. Right channel will be under 70 dB (typical is 75 dB). Correct the phase control of the Belar to improve this figure. Calibrate channel separation to the minimum Right level, using **C1** trimmer at the low pass filter of SC100. More than 70 dB rejection is a correct figure. If not, make the complete procedure of channel separation calibration of the Belar set (see Belar manual).

f) Now, connect the sine wave generator, at the same frequency and level, to the Right Channel (pin **JS3**).

g) Measure at the Belar the residual level of the Left channel. Calibrate channel separation to the minimum Left level, using **P2** control. More than 70 dB rejection is a correct figure.

h) Verify channel separation at the 20 Hz – 15.000 Hz range. This must be better than 60 dB (65 dB typical).

7.2.5 MEASUREMENT OF RESIDUAL NOISE

Disconnect **S1** and **S3** from the Audio Processor. Use a good Audio Voltmeter with A weighted filter connected at the Left and Right audio outputs of the BELAR modulation monitor. Use as reference 4 Vpp sine wave. Values better than 92 dBA must be measured (94 dBA is typical value)

7.2.6 AUDIO DISTORTION MEASUREMENTS

Connect a good distortion measurement set like System ONE, Agilent 8903B or Sound Tech ST 1710A to the Left & Right audio outputs of the BELAR set. Values under 0,01 % must be measured at 1 kHz.

Since Belar has it distortion floor at 0,01%, in order to measure the real distortion of SC100 stereo coder, is recommended to use a procedure not based on modulation monitors. Proceed this way:

a) Connect to ground Test Point **TP1** in order to cancel 19 KHz Pilot

b) Then, connect a very low distortion audio generator at both inputs **S1** & **S3**. This will cancel 38 Khz subcarrier. Connect a good THD distortion measurement set at the MPX output of stereo coder. Adjust to minimum value the THD meter.

c) Use the monitor output of the THD measurement set to analyze the distortion products in order to separate 38 KHz residual and noise from the true distortion components. Use a Tektronix 5L4 N analyzer, or TiePie FFT HS3 probe, or SoundTech Lab software, etc. Identify the harmonics of 1 KHz and calculate:

$$Distorsión = \frac{100}{H1} \times \left(H2^2 + H3^2 + H4^2 + \text{h}\right)^{1/2}$$

H1 = Level of fundamental tone.

Hn = Level of harmonics (reduced by the gain set of the THD meter).

d) Verify that the distortion level is lower than 0,003 % at 1 KHz.

Chapter 8	Technical specifications

Analog Input	600/10 K balanced XLR 50 dB CM Rejection 20-15 Khz. Input level selected by software in 1 dB steps. Sigma-Delta converters 24 bits / 192 Khz
Digital Input	Optional AES-3 digital balanced input Z=110 ohms. Automatic selection of 32, 44,1, 48, 96 & 192 Khz with sample rate converter (128 dB Dynamic Range, -117 dB THD) to avoid jitter
Analog Output	600 balanced XLR, output level +4 dBm Sigma-Delta converters 24 bits / 192 Khz
Digital Output	Optional AES-3 digital balanced output, Z=110 ohms FS=48 or 96 Khz, selected by software
MPX Output	From 2 Vpp to 5 Vpp in 0.2 steps
Processing Technology	DSP (Digital Signal Processing). Total CPU power 2.700 MIPS
MPX Post-Processing	SuperModulation exclusive Solidyne technology, at 608 Khz oversampling. Fast clipper DC-10 Mhz wideband channel to avoid audible *artifacts*
Frequency Response	20-15.000 Hz +/-0.25 dB. Flat mode XLR out or digital AES-3 Out. Output without pre-emphasis
Harmonic Distortion (THD)	THD below 0.008 % (30-15 Khz, Flat Mode)
Noise	Dynamic Range= 95 dBA
Stereo Separation	75 dBA @ 1 KHz: > 65 dBA @ 30-15.000 Hz
Subsonic Filter	Chebyshev FC=20 Hz 25 dB rejection at 10 Hz
Asymmetry Canceling	Phase processing technology with Kahn-Bonello algorithms. Cancelling Factor=8:1
Linear Expander	Range=20 dB Attack & Release time software controlled
Gated Wide-Band AGC	Range=30 dB. Attack / Release time & Threshold controlled from LCD screen
Multiband Compressors	DSP controlled. Five Bands, Crossover= 24dB/oct Max compression = 30dB. Slope = 10:1
Compressors Attack/Release Time	Attack and Release controlled separately
IM Canceled Clipper	IM canceling factor : greater than 30 dB below 250 Hz
Fast Clipper	Four channels, absolutely alias free using DC-10 Mhz bandwidth channel
Density Equalizer	Four bands density equalizer with 15 dB range at the output of multiband compressors
Low pass filter	15 Khz digital low pass FIR filter, 60 dB rejection at 19 Khz
Storage of Preset Settings	30 programs that can be changed on-air from PC computer using the 462dsp serial port
RS-232 PC control	RS232 serial port. It can be connected to USB bus with optional external adapter. Optional Ethernet bus connection
Super Modulation	MPX processing for stereo interleaving, allows for 130% L & R audio level at 100% modulation
RS-232 PC control	Yes. It includes free Windows 2000/XP software
AM FM compatible	The user can change the processor's AM/FM firmware chips. Only two socket mounted Flash EPROM IC's have to be replaced. Stereo coder is a separate module
Processing Bands and stages	Five processing bands / Nine processing stages
LCD Display	Blue color, LCD display with backlight. Graphic type Resolution 240 x 64
RDS Encoder	Optional built in RDS encoder
Power	90/127V and 190/230V; 50/60 Hz, selectable from rear panel
Dimensions	483 mm Wide 240 mm Deep, 88 mm High

SC-100 Digital Stereo Generator
16x Oversampling - Very Low Distortion

Audio Input Impedance	600/5 kOhms
Audio Input Level	1,5 V rms for 5 Vpp at MPX out @ 400 Hz
MPX Output	Differential output, BNC connector, floating ground 50 ohms Allows 45 dB canceling buzz & noise due to ground loops
Composite Output Level	2 - 5.5 Volts pp, adjustable from LCD display
Frequency Response	20-14.000 Hz +/- 0,1 dB Elliptical low pass filter; -1 dB at 15 kHz / -60 dB at 19 kHz
Audio Input Filtering	15 kHz, active FDNR filter 5 poles, elliptical
Total Distortion	0.003 % at 1 kHz.
Signal to Noise Ratio	95 dBA or better, Ref 100% modulation.
Stereo Separation	75 dB at 400 Hz / > 65 dB; 30-15.000 Hz.
Crosstalk	Main to sub & sub to main due to amplitude and phase nonlinearities of left and right channels, 30-15.000 Hz; 65 dB minimum, below 100% modulation.
38 kHz Suppression	75 dB minimum below 100% modulation.
57, 76 and 95 kHz Suppression	75 dB minimum below 100% modulation.
76 kHz Sideband Suppression	75 dB minimum below 100% modulation.
Pilot Level	Adjusted 7-12 % from rear panel preset control
Pilot Protection	70 dB at 19 Khz
RDS channel protection	> 60 dB
Pilot Stability	+/- 0.05 Hz, 0 to 50 °C.